WHAT IS CLAIMED IS:

1. A method for low-frequency emphasizing the spectrum of a sound signal transformed in a frequency domain and comprising transform coefficients grouped in a number of blocks, comprising:

calculating a maximum energy for one block having a position index;

calculating a factor for each block having a position index smaller than the position index of the block with maximum energy, the calculation of a factor comprising, for each block:

- computing an energy of the block; and
- computing the factor from the calculated maximum energy and the computed energy of the block; and

for each block, determining from the factor a gain applied to the transform coefficients of the block.

2. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, wherein the transform coefficients are Fast Fourier Transform coefficients.

3. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, comprising applying an adaptive low-frequency emphasis to the spectrum of the sound signal to minimize a perceived distortion in lower frequencies of the spectrum.

4. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, comprising grouping the transform coefficients in blocks of a predetermined number of consecutive transform coefficients.

5. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, wherein:

- calculating a maximum energy for one block comprises:

computing the energy of each block up to a given position in the spectrum; and

storing the energy of the block with maximum energy; and

- determining a position index comprises:

storing the position index of the block with maximum energy.

6. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 5, wherein computing the energy of each block up to a given position in the spectrum comprises:

computing the energy of each block up to the first quarter of the spectrum.

7. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, wherein computing the factor for each block comprises:

computing a ratio R_m for each block with a position index m smaller than the position index of the block with maximum energy, using the relation

$$R_m = E_{max} / E_m$$

where E_{max} is the calculated maximum energy and E_m the computed energy for block corresponding to position index m.

8. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 7, comprising setting the ratio R_m to a predetermined value when R_m is larger than said predetermined value.

9. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 7, comprising setting the ratio $R_m = R_{(m-1)}$ when $R_m > R_{(m-1)}$.

10. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, wherein computing the factor comprises setting the

factor to a predetermined value when the factor is larger than said predetermined value.

11. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 1, wherein computing the factor comprises setting the factor for one block to the factor of the preceding block when the factor of said one block is larger than the factor of the preceding block.

12. A method for low-frequency emphasizing the spectrum of a sound signal as defined in claim 7, wherein computing the factor further comprises calculating a value $(R_m)^{1/4}$, and applying the value $(R_m)^{1/4}$ as a gain for the transform coefficient of the corresponding block.

13. A device for low-frequency emphasizing the spectrum of a sound signal transformed in a frequency domain and comprising transform coefficients grouped in a number of blocks, comprising:

means for calculating a maximum energy for one block having a position index;

means for calculating a factor for each block having a position index smaller than the position index of the block with maximum energy, the factor calculating means comprising, for each block:

- means for computing an energy of the block; and
- means for computing the factor from the calculated maximum energy and the computed energy of the block; and

means for determining, for each block and from the factor, a gain applied to the transform coefficients of the block.

14. A device for low-frequency emphasizing the spectrum of a sound signal transformed in a frequency domain and comprising transform coefficients grouped in a number of blocks, comprising:

a calculator of a maximum energy for one block having a position index;

a calculator of a factor for each block having a position index smaller than the position index of the block with maximum energy, wherein the factor calculator, for each block:

- computes an energy of the block; and
- computes the factor from the calculated maximum energy and the computed energy of the block; and

a calculator of a gain, for each block and in response to the factor, the gain being applied to the transform coefficients of the block.

15. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 14, wherein the transform coefficients are Fast Fourier Transform coefficients.

16. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 14, wherein the transform coefficients are grouped in blocks of a predetermined number of consecutive transform coefficients.

17. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 14, wherein the maximum energy calculator:

computes the energy of each block up to a predetermined position in the spectrum; and

comprises a store for the maximum energy; and

comprises a store for the position index of the block with maximum energy.

18. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 17, wherein the maximum energy calculator computes the energy of each block up to the first quarter of the spectrum.

19. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 14, wherein the factor calculator:

computes a ratio R_m for each block with a position index m smaller than the position index of the block with maximum energy, using the relation

$$R_m = E_{max} / E_m$$

where E_{max} is the calculated maximum energy and E_m the computed energy for the block corresponding to the position index m.

20. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 19, wherein the factor calculator sets the ratio R_m to a predetermined value when R_m is larger than said predetermined value.

21. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 19, wherein the factor calculator sets the ratio the ratio $R_m = R_{(m-1)}$ when $R_m > R_{(m-1)}$.

22. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 14, wherein the factor calculator sets the factor to a predetermined value when the factor is larger than said predetermined value.

23. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 14, wherein the factor calculator sets the factor for one block to the factor of the preceding block when the factor of said one block is larger than the factor of the preceding block.

24. A device for low-frequency emphasizing the spectrum of a sound signal as defined in claim 19, wherein:

the factor calculator computes a value $(R_m)^{1/4}$; and

the gain calculator applies the value $(R_m)^{1/4}$ as a gain for the transform coefficient of the corresponding block.

25. A method for processing a received, coded sound signal comprising:

extracting coding parameters from the received, coded sound signal, the extracted coding parameters including transform coefficients of a frequency transform of said sound signal, wherein the transform coefficients were low-frequency emphasized using a method as defined in any of claims 1 to 12;

processing the extracted coding parameters to synthesize the sound signal, processing the extracted coding parameters comprising low-frequency de-emphasizing the low-frequency emphasized transform coefficients.

26. A method for processing a received, coded sound signal as defined in claim 25, wherein:

extracting coding parameters comprises dividing the low-frequency emphasized transform coefficients into a number K of blocks of transform coefficients; and

low-frequency de-emphasizing the low-frequency emphasized transform coefficients comprises scaling the transform coefficients of at least a portion of the K blocks to cancel the low-frequency emphasis of the transform coefficients.

27. A method for processing a received, coded sound signal as defined in claim 26, wherein:

low-frequency de-emphasizing the low-frequency emphasized transform coefficients comprises scaling the transform coefficients of the first K/s blocks of said K blocks of transform coefficients, s being an integer.

28. A method for processing a received, coded sound signal as defined in claim 27, wherein scaling the transform coefficients comprises:

computing the energy ε_k of each of the K blocks of transform coefficients;

computing the maximum energy ε_{max} of one block amongst the first K/s blocks; and

computing for each of the first K/s blocks a factor fac_k; and

scaling the transform coefficients of each of the first K/s blocks using the factor fac_k of the corresponding block.

29. A method for processing a received, coded sound signal as defined in claim 28, wherein computing for each of the first K/s blocks, up to a position index of the block with maximum energy, a factor fac_k comprises using the following expressions:

$$fac_0 = \max((\varepsilon_0 / \varepsilon_{max})^{0.5}; 0.1)$$

$fac_k = \max((\varepsilon_k / \varepsilon_{max})^{0.5}, fac_{k-1})$ for $k=1,\ldots, K/s-1$, where ε_k is the energy of the block with index k.

30. A decoder for processing a received, coded sound signal comprising:

an input decoder portion supplied with the received, coded sound signal and implementing an extractor of coding parameters from the received, coded sound signal, the extracted coding parameters including transform coefficients of a frequency transform of said sound signal, wherein the transform coefficients were low-frequency emphasized using a device as defined in any of claims 13 to 24;

a processor of the extracted coding parameters to synthesize the sound signal, said processor comprising a low-frequency de-emphasis module supplied with the low-frequency emphasized transform coefficients.

31. A decoder as defined in claim 30, wherein:

the extractor divides the low-frequency emphasized transform coefficients into a number K of blocks of transform coefficients; and

the low-frequency de-emphasis module scales the transform coefficients of at least a portion of the K blocks to cancel the low-frequency emphasis of the transform coefficients.

32. A decoder as defined in claim 31, wherein:

the low-frequency de-emphasis module scales the transform coefficients of the first K/s blocks of said K blocks of transform coefficients, s being an integer.

33. A decoder as defined in claim 32, wherein the low-frequency de-emphasis module:

computes the energy ε_k of each of the K/s blocks of transform coefficients;

computes the maximum energy ε_{max} of one block amongst the first K/s blocks; and

computes for each of the first K/s blocks a factor fac_k; and

scales the transform coefficients of each of the first K/s blocks using the factor fac_k of the corresponding block.

34. A decoder as defined in claim 33, wherein the low-frequency de-emphasis module calculates the factor fac_k using the following expressions:

$$\mathrm{fac}_0 = \max\left((\varepsilon_0 / \varepsilon_{max})^{0.5}, 0.1\right)$$

$$\mathrm{fac}_k = \max\left((\varepsilon_k / \varepsilon_{max})^{0.5}, \mathrm{fac}_{k-1}\right) \text{ for } k=1,\ldots, K/s-1,$$

where ε_k is the energy of the block with index k.

35. An HF coding method for coding, through a bandwidth extension scheme, an HF signal obtained from separation of a full-bandwidth sound signal into the HF signal and a LF signal, comprising:

performing an LPC analysis on the LF and HF signals to produce LPC coefficients which model a spectral envelope of the LF and HF signals;

calculating, from the LPC coefficients, an estimation of an HF matching gain;

calculating the energy of the HF signal;

processing the LF signal to produce a synthesized version of the HF signal;

calculating the energy of the synthesized version of the HF signal;

calculating a ratio between the calculated energy of the HF signal and the calculated energy of the synthesized version of the HF signal, and expressing the calculated ratio as an HF compensating gain; and

calculating a difference between the estimation of the HF matching gain and the HF compensating gain to obtain a gain correction;

wherein the coded HF signal comprises the LPC parameters and the gain correction.

36. An HF coding method as defined in claim 35, wherein the HF signal is composed of frequency components higher than 6400 Hz.

37. An HF coding method as defined in claim 35, further comprising:
converting the LPC coefficients to ISF coefficients; and
quantizing the ISF coefficients for transmission.

38. An HF coding method as defined in claim 37, further comprising:
converting the quantized ISF coefficients to quantized ISP coefficients; and
converting the quantized ISP coefficients to quantized LPC coefficients.

39. An HF coding method as defined in claim 35, wherein processing the LF signal to produce a synthesized version of the HF signal comprises:

filtering the LF signal through a quantized version of a LPC filter which models a spectral envelope of the HF signal to produce a residual signal; and

filtering the residual signal through a quantized HF synthesis filter to produce the synthesized version of the HF signal.

40. An HF coding method as defined in claim 35, wherein:

- calculating the energy of the HF signal comprises:

filtering the HF signal through a HF perceptual filter; and

calculating the energy of the perceptually filtered HF signal; and

- calculating the energy of the synthesized version of the HF signal comprises:

filtering the synthesized version of the HF signal through a HF perceptual filter; and

calculating the energy of the perceptually filtered synthesized version of the HF signal.

41. An HF coding method as defined in claim 35, wherein expressing the calculated ratio as a HF gain comprises:

expressing in dB the calculated ratio between the calculated energy of the HF signal and the calculated energy of the synthesized version of the HF signal.

41a. An HF coding method as defined in claim 35, wherein calculating the HF matching gain comprises computing a ratio between the frequency responses of the LF LPC filter and the HF LPC filter at the Nyquist frequency.

42. An HF coding method as defined in claim 35, wherein:

- performing an LPC analysis comprises computing HF quantized LPC coefficients $\hat{A}_{HF}(z)$; and

- calculating an estimation of an HF matching gain comprises:

computing 64 samples of a decaying sinusoid *h(n)* at Nyquist frequency per sample by filtering a unit impulse $\delta(n)$ through a one-pole filter of the form $1/(1+0.9z^{-1})$;

filtering the decaying sinusoid *h(n)* through a LF LPC filter $\hat{A}(z)$ to obtain a low-frequency residual, wherein $\hat{A}(z)$ represents LF quantized LPC coefficients from a LF coder;

filtering the filtered decaying sinusoid $h(n)$ through an HF LPC synthesis filter $1/\hat{A}_{HF}(z)$ to obtain a synthesis signal $x(n)$; and

computing a multiplicative inverse of the energy of the synthesis signal $x(n)$, and expressing it in the logarithmic domain, to produce a gain g_{match}; and

interpolating the gain g_{match} to produce the estimation of the HF matching gain.

43. An HF coding method as defined in claim 35, comprising quantizing the gain correction to obtain a quantized gain correction.

44. An HF coding device for coding, through a bandwidth extension scheme, an HF signal obtained from separation of a full-bandwidth sound signal into the HF signal and a LF signal, comprising:

means for performing an LPC analysis on the LF and HF signals to produce LPC coefficients which model a spectral envelope of the LF and HF signals;

means for calculating, from the LPC coefficients, an estimation of an HF matching gain;

means for calculating the energy of the HF signal;

means for processing the LF signal to produce a synthesized version of the HF signal;

means for calculating the energy of the synthesized version of the HF signal;

means for calculating a ratio between the calculated energy of the HF signal and the calculated energy of the synthesized version of the HF signal, and means for expressing the calculated ratio as an HF compensating gain; and

means for calculating a difference between the estimation of the HF matching gain and the HF compensating gain to obtain a gain correction;

wherein the coded HF signal comprises the LPC parameters and the gain correction.

45. An HF coding device for coding, through a bandwidth extension scheme, an HF signal obtained from separation of a full-bandwidth sound signal into the HF signal and a LF signal, comprising:

an LPC analyzing means supplied with the LF and HF signals and producing, in response to the HF signal, LPC coefficients which model a spectral envelope of the LF and HF signals;

a calculator of an estimation of an matching HF gain in response to the LPC coefficients;

a calculator of the energy of the HF signal;

a filter supplied with the LF signal and producing, in response to the LF signal, a synthesized version of the HF signal;

a calculator of the energy of the synthesized version of the HF signal;

a calculator of a ratio between the calculated energy of the HF signal and the calculated energy of the synthesized version of the HF signal;

a converter supplied with the calculated ratio and expressing said calculated ratio as an HF compensating gain; and

a calculator of a difference between the estimation of the HF matching gain and the HF compensating gain to obtain a gain correction;

wherein the coded HF signal comprises the LPC parameters and the gain correction.

46. An HF coding device as defined in claim 45, wherein the HF signal is composed of frequency components higher than 6400 Hz.

47. An HF coding device as defined in claim 45, further comprising:

a converter of the LPC coefficients to ISF coefficients; and

a quantizer of the ISF coefficients.

48. An HF coding device as defined in claim 47, further comprising:

a converter of the quantized ISF coefficients to quantized ISP coefficients; and

a converter of the quantized ISP coefficients to quantized LPC coefficients.

49. An HF coding device as defined in claim 45, wherein the filter supplied with the LF signal and producing, in response to the LF signal, a synthesized version of the HF signal comprises:

a quantized LPC filter supplied with the LF signal and producing, in response to the LF signal, a residual signal; and

a quantized HF synthesis filter supplied with the residual signal and producing, in response to the residual signal, the synthesized version of the HF signal.

50. An HF coding device as defined in claim 45, wherein:

- the calculator of the energy of the HF signal comprise:

a HF perceptual filter supplied with the HF signal; and

a calculator of the energy of the perceptually filtered HF signal; and

- the calculator of the energy of the synthesized version of the HF signal comprises:

a HF perceptual filter supplied with the synthesized version of the HF signal; and

a calculator of the energy of the perceptually filtered synthesized version of the HF signal.

51. An HF coding device as defined in claim 45, wherein the converter expressing the calculated ratio as a HF gain comprises:

means for expressing in dB the calculated ratio between the calculated energy of the HF signal and the calculated energy of the synthesized version of the HF signal.

51a. An HF coding device as defined in claim 55, wherein the calculator of the HF matching gain computes a ratio between the frequency responses of the LF LPC filter and the HF LPC filter at the Nyquist frequency.

52. An HF coding device as defined in claim 45, wherein:

- the LPC analyzer comprises a calculator of HF quantized LPC coefficients $\hat{A}_{HF}(z)$; and
- the calculator of an estimation of an HF matching gain comprises:

a calculator of 64 samples of a decaying sinusoid $h(n)$ at Nyquist frequency π radians per sample by filtering a unit impulse $\delta(n)$ through a one-pole filter of the form $1/(1+0.9z^{-1})$;

a LF LPC filter $\hat{A}(z)$ for filtering the decaying sinusoid $h(n)$ to obtain a low-frequency residual, wherein $\hat{A}(z)$ represents LF quantized LPC coefficients from a LF coder;

an HF LPC synthesis filter $1/\hat{A}_{HF}(z)$ for filtering the filtered decaying sinusoid $h(n)$ to obtain a synthesis signal $x(n)$; and

a calculator of a multiplicative inverse of the energy of the synthesis signal $x(n)$, and expressing it in the logarithmic domain, to produce a gain g_{match}; and

an interpolator of the gain g_{match} to produce the estimation of the HF matching gain.

53. An HF coding device as defined in claim 45, comprising a quantizer of the gain correction to obtain a quantized gain correction.

54. A method for decoding an HF signal coded through a bandwidth extension scheme, comprising:

receiving the coded HF signal;

extracting from the coded HF signal LPC coefficients and a gain correction;

calculating an estimation of the HF gain from the extracted LPC coefficients;

adding the gain correction to the calculated estimation of the HF gain to obtain an HF gain;

amplifying a LF excitation signal by the HF gain to produce a HF excitation signal; and

processing the HF excitation signal through a HF synthesis filter to produce a synthesized version of the HF signal.

55. A method for decoding an HF signal as defined in claim 54, further comprising reducing buzziness of the HF excitation signal before supplying said HF excitation signal to the HF synthesis filter.

56. A method for decoding an HF signal as defined in claim 54, wherein the HF synthesis filter is a HF linear-predictive synthesis filter.

57. A method for decoding an HF signal as defined in claim 54, further comprising HF energy smoothing the synthesized version of the HF signal to smooth energy variations in said synthesized version of the HF signal.

58. A method for decoding an HF signal as defined in claim 54, wherein extracting from the coded HF signal the LPC coefficients comprises:

decoding ISF coefficients from the coded HF signal;

converting the ISF coefficients to ISP coefficients;

interpolating the ISP coefficients; and

converting the interpolated ISP coefficients to quantized HF LPC coefficients.

59. A method for decoding an HF signal as defined in claim 54, wherein:

- extracting LPC coefficients comprises extracting from the coded HF signal HF quantized LPC coefficients $\hat{A}_{HF}(z)$; and

- calculating an estimation of a HF gain comprises:

computing from the extracted LPC parameters ;

computing 64 samples of a decaying sinusoid $h(n)$ at Nyquist frequency π radians per sample by filtering a unit impulse $\delta(n)$ through a one-pole filter of the form $1/(1+0.9z^{-1})$;

filtering the decaying sinusoid $h(n)$ through a LF LPC filter $\hat{A}(z)$ to obtain a low-frequency residual, wherein $\hat{A}(z)$ represents LF quantized LPC coefficients from a LF decoder;

filtering the filtered decaying sinusoid $h(n)$ through an HF LPC synthesis filter $1/\hat{A}_{HF}(z)$ to obtain a synthesis signal $x(n)$; and

computing a multiplicative inverse of the energy of the synthesis signal $x(n)$, and expressing it in the logarithmic domain, to produce a gain g_{match}; and

interpolating the gain g_{match} to produce the estimation of the HF gain.

60. A decoder for decoding an HF signal coded through a bandwidth extension scheme, comprising:

means for receiving the coded HF signal;

means for extracting from the coded HF signal LPC coefficients and a gain correction;

means for calculating an estimation of the HF gain from the extracted LPC coefficients;

means for adding the gain correction to the calculated estimation of the HF gain to obtain an HF gain;

means for amplifying a LF excitation signal by the HF gain to produce a HF excitation signal; and

means for processing the HF excitation signal through a HF synthesis filter to produce a synthesized version of the HF signal.

61. A decoder for decoding an HF signal coded through a bandwidth extension scheme, comprising:

an input for receiving the coded HF signal;

a decoder supplied with the coded HF signal and extracting from the coded HF signal LPC coefficients;

a decoder supplied with the coded HF signal and extracting from the coded HF signal a gain correction;

a calculator of an estimation of the HF gain from the extracted LPC coefficients;

an adder of the gain correction and the calculated estimation of the HF gain to obtain an HF gain;

an amplifier of a LF excitation signal by the HF gain to produce a HF excitation signal; and

a HF synthesis filter supplied with the HF excitation signal and producing, in response to the HF excitation signal, a synthesized version of the HF signal.

62. A decoder for decoding an HF signal as defined in claim 61, further comprising a buzziness reducer supplied with the HF excitation signal before supplying said HF excitation signal to the HF synthesis filter.

63. A decoder for decoding an HF signal as defined in claim 61, wherein the HF synthesis filter is a HF linear-predictive synthesis filter.

64. A decoder for decoding an HF signal as defined in claim 61, further comprising an HF energy smoothing module supplied with the synthesized version of the HF signal, the HF energy smoothing module smoothing energy variations in the synthesized version of the HF signal.

65. A decoder for decoding an HF signal as defined in claim 61, wherein the decoder extracting from the coded HF signal the LPC coefficients comprises:

a decoder of ISF coefficients from the coded HF signal;

a converter the ISF coefficients to ISP coefficients;

an interpolator of the ISP coefficients; and

a converter of the interpolated ISP coefficients to quantized HF LPC coefficients.

66. A decoder for decoding an HF signal as defined in claim 61, wherein:

- the decoder extracting LPC coefficients comprises an extractor of quantized LPC coefficients $\hat{A}_{HF}(z)$ from the coded HF signal; and

- the calculator of an estimation of the HF gain comprises:

a calculator of 64 samples of a decaying sinusoid $h(n)$ at Nyquist frequency π radians per sample by filtering a unit impulse $\delta(n)$ through a one-pole filter of the form $1/(1+0.9z^{-1})$;

a LF LPC filter $\hat{A}(z)$ for filtering the decaying sinusoid $h(n)$ to obtain a low-frequency residual, wherein $\hat{A}(z)$ represents LF quantized LPC coefficients from a LF decoder;

an HF LPC synthesis filter $1/\hat{A}_{HF}(z)$ for filtering the filtered decaying sinusoid $h(n)$ to obtain a synthesis signal $x(n)$; and

a calculator of a multiplicative inverse of the energy of the synthesis signal $x(n)$, and expressing it in the logarithmic domain, to produce a gain g_{match}; and

an interpolator of the gain g_{match} to produce the estimation of the HF gain.

67. A method of switching from a first sound signal coding mode to a second sound signal coding mode at the junction between a previous frame coded according to the first coding mode and a current frame coded according to the second coding mode, wherein the sound signal is filtered through a weighting filter to produce, in the current frame, a weighted signal, comprising:

calculating a zero-input response of the weighting filter;

windowing the zero-input response so that said zero-input response has an amplitude monotonically decreasing to zero after a predetermined time period; and

in the current frame, removing from the weighted signal the windowed zero-input response.

68. A method of switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 67, wherein calculating a zero-input response of the weighting filter comprises calculating a zero-input response in the weighted domain.

69. A method of switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 67, wherein the first coding mode is an ACELP coding mode and the second coding mode is a TCX coding mode.

70. A method of switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 67, wherein windowing the zero-input response comprises truncating said zero-input response to the predetermined time period.

71. A method of switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 67, comprising, after the windowed zero-input response has been removed from the weighted signal, windowing the weighted signal into a TCX frame of predetermined duration.

72. A method of switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 71, further comprising transforming into the frequency domain the weighted signal windowed into a TCX frame of predetermined duration.

73. A method of switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 67, wherein the weighting filter is a perceptual weighting filter.

74. A device for switching from a first sound signal coding mode to a second sound signal coding mode at the junction between a previous frame coded according to the first coding mode and a current frame coded according to the second coding mode, wherein the sound signal is filtered through a weighting filter to produce, in the current frame, a weighted signal, comprising:

means for calculating a zero-input response of the weighting filter;

means for windowing the zero-input response so that said zero-input response has an amplitude monotonically decreasing to zero after a predetermined time period; and

means for removing, in the current frame, the windowed zero-input response from the weighted signal.

75. A device for switching from a first sound signal coding mode to a second sound signal coding mode at the junction between a previous frame coded according to the first coding mode and a current frame coded according to the second coding mode, wherein the sound signal is filtered through a weighting filter to produce, in the current frame, a weighted signal, comprising:

a calculator of a zero-input response of the weighting filter;

a window generator for windowing the zero-input response so that said zero-input response has an amplitude monotonically decreasing to zero after a predetermined time period; and

an adder for removing, in the current frame, the windowed zero-input response from the weighted signal.

76. A device for switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 75, wherein the zero-input response calculator calculates a zero-input response in the weighted domain.

77. A device for switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 75, wherein the first coding mode is an ACELP coding mode and the second coding mode is a TCX coding mode.

78. A device for switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 75, wherein the window generator truncates the zero-input response to the predetermined time period.

79. A device for switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 75, comprising another window generator for windowing, after the windowed zero-input response has been removed from the weighted signal, the weighted signal into a TCX frame of predetermined duration.

80. A device for switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 79, further comprising a frequency transform module which, in operation, transforms in the frequency domain the weighted signal windowed into a TCX frame of predetermined duration.

81. A device for switching from a first sound signal coding mode to a second sound signal coding mode as defined in claim 67, wherein the weighting filter is a perceptual weighting filter.

82. A method for producing from a decoded target signal an overlap-add target signal in a current frame coded according to a first coding mode, comprising:

windowing the decoded target signal of the current frame in a given window;

skipping a left portion of the window;

calculating a zero-input response of a weighting filter of the previous frame coded according to a second coding mode, and windowing the zero-input response so that said zero-input response has an amplitude monotonically decreasing to zero after a predetermined time period; and

adding the calculated zero-input response to the decoded target signal to reconstruct said overlap-add target signal.

83. A method for producing an overlap-add target signal as defined in claim 82, comprising weighting the calculated zero-input response prior to windowing said calculated zero-input response.

84. A method for producing an overlap-add target signal as defined in claim 83, wherein weighting the calculated zero-input response comprises perceptually weighting said calculated zero-input response.

85. A method for producing an overlap-add target signal as defined in claim 82, comprising saving in a buffer a last portion of samples of the current frame.

86. A method for producing an overlap-add target signal as defined in claim 82, wherein the windowed, calculated zero-input response has an amplitude monotonically decreasing to zero after 10 ms.

87. A device for producing from a decoded target signal an overlap-add target signal in a current frame coded according to a first coding mode, comprising:

means for windowing the decoded target signal of the current frame in a given window;

means for skipping a left portion of the window;

means for calculating a zero-input response of a weighting filter of the previous frame coded according to a second coding mode, and means for windowing the zero-input response so that said zero-input response has an amplitude monotonically decreasing to zero after a predetermined time period; and

means for adding the calculated zero-input response to the decoded target signal to reconstruct said overlap-add target signal.

88. A device for producing from a decoded target signal an overlap-add target signal in a current frame coded according to a first coding mode, comprising:

a first window generator for windowing the decoded target signal of the current frame in a given window;

means for skipping a left portion of the window;

a calculator of a zero-input response of a weighting filter of the previous frame coded according to a second coding mode, and a second window generator for windowing the zero-input response so that said zero-input response has an amplitude monotonically decreasing to zero after a predetermined time period; and

an adder for adding the calculated zero-input response to the decoded target signal to reconstruct said overlap-add target signal.

89. A device for producing an overlap-add target signal as defined in claim 88, comprising a filter for weighting the calculated zero-input response prior to windowing said calculated zero-input response.

90. A device for producing an overlap-add target signal as defined in claim 89, wherein the weighting filter is a perceptual weighting filter.

91. A device for producing an overlap-add target signal as defined in claim 88, comprising a buffer for saving a last portion of samples of the current frame.

92. A device for producing an overlap-add target signal as defined in claim 88, wherein the windowed, calculated zero-input response has an amplitude monotonically decreasing to zero after 10 ms.